CLAIMS

What is claimed is:

1. In a system having a working memory and a digital processor, a method for encoding speech signals, comprising:

   providing an encoder including (a) a pitch predictor and (b) a source excitation codebook, the pitch predictor having various parameters and being a multi-tap pitch predictor utilizing a codebook subdivided into at least a first vector codebook and a second vector codebook;

   using the pitch predictor, (i) removing certain redundancies in a subject speech signal and (ii) vector quantizing the pitch predictor parameters; and

   using the source excitation codebook, indicating pulses in the subject speech signal by deriving corresponding vector values.

2. The method as claimed in Claim 1 wherein deriving corresponding vector values is an open-loop derivation.

3. The method as claimed in Claim 2 wherein the open-looped derivation is complete in a single-pass.

4. The method as claimed in Claim 1 wherein the pulses are represented by ternary values (1, 0, -1).

5. The method as claimed in Claim 1 wherein the vector quantizing is product code vector quantizing.

6. The method as claimed in Claim 1 wherein the pitch predictor codebook is optimized in a closed-loop manner.

7. The method as claimed in Claim 1 wherein the pitch predictor codebook is optimized then the source excitation codebook is optimized.

8. In a system having a working memory and a digital processor, anapparatus for encoding speech signals comprising:

a pitch predictor to remove certain redundancies in a subject speech signal, the pitch predictor having vector quantized parameters and being a multi-tap pitch predictor utilizing a codebook subdivided into at least a first vector codebook and a second vector codebook; and

a source excitation codebook coupled to receive speech signals from the pitch predictor, the source excitation codebook indicating pulses in the subject speech signal by deriving corresponding vector values.

9. The apparatus as claimed in Claim 8 wherein the vector values are derived in an open-loop manner.

10. The apparatus as claimed in Claim 9 wherein the open-loop manner is complete in a single-pass.

11. The apparatus as claimed in Claim 8 wherein the pulses are represented by ternary values (1, 0, -1).

12. The apparatus as claimed in Claim 8 wherein the vector quantized parameters are quantized using product code vector quantization.

13. The apparatus as claimed in Claim 8 wherein the pitch predictor codebook is optimized in a closed-loop manner.

14. The apparatus as claimed in Claim 8 wherein the pitch predictor codebook is optimized then the source excitation codebook is optimized.

15. A system for encoding speech signals, comprising:

an electronic device having a working memory and a digital processor;

an encoder executable in the working memory by the digital processor, the encoder including:

a pitch predictor to remove certain redundancies in a subject speech signal, the pitch predictor having vector quantized parameters and

being a multi-tap pitch predictor utilizing a codebook subdivided into at least a first vector codebook and a second vector codebook; and

a source excitation codebook coupled to receive speech signals from the pitch predictor, the source excitation codebook indicating pulses in the subject speech signal by deriving corresponding vector values.

16. The system as claimed in Claim 15 wherein the corresponding vector values are derived in an open-loop manner.

17. The system as claimed in Claim 16 wherein the open-loop manner is complete in a single-pass.

18. The system as claimed in Claim 15 wherein the pulses are represented by ternary values (1, 0, -1).

19. The system as claimed in Claim 15 wherein the vector quantized parameters are quantized using product code vector quantization.

20. The system as claimed in Claim 15 wherein the pitch predictor codebook is optimized in a closed-loop manner.

21. The system as claimed in Claim 15 wherein the pitch predictor codebook is optimized then the source excitation codebook is optimized.

22. The system as claimed in Claim 15 wherein the electronic device is a personal communication device.

23. The system as claimed in Claim 22 wherein the personal communication device is selected from a group consisting of secure telephones, cellular phones, answering machines, voicemail, and digital memorandum recorders.

24. In a system having working memory and a digital processor, a method for performing multi-tap pitch predictor vector quantization, the method comprising:

providing an adaptive codebook;

providing at least one pitch predictor codebook having predictor coefficients; and

adjusting the adaptive codebook with a contribution from the adaptive codebook in combination with the predictor coefficients, the predictor coefficients being selected by searching the at least one pitch predictor codebook.

25. The method as claimed in Claim 24 further including filtering the combination and computing an error signal between a target speech signal and the filtered combination.

26. The method as claimed in Claim 25 wherein the searching is a function of the error signal.

27. The method as claimed in Claim 25 wherein the filtering is weighted synthesis filtering.

28. The method as claimed in Claim 25 wherein adjusting the adaptive codebook includes adjusting a lag factor.

29. The method as claimed in Claim 28 wherein the lag factor is a function of the error signal.

30. The method as claimed in Claim 24 wherein the vector quantization is conventional vector quantization.

31. The method as claimed in Claim 24 wherein the vector quantization is product code vector quantization.

32. The method as claimed in Claim 24 wherein the searching includes linear predictive analysis-by-synthesis searching.

33. In a system having working memory and a digital processor, a multi-tap pitch predictor for performing vector quantization, comprising:

    at least one pitch predictor codebook having predictor coefficients; and

    an adaptive codebook adjusted with a contribution from the adaptive codebook in combination with the predictor coefficients, the predictor coefficients being selected by searching the at least one pitch predictor codebook.

34. The pitch predictor as claimed in Claim 33 further including a filter to filter the combination and compute an error signal between a target speech signal and the output of the filter.

35. The pitch predictor as claimed in Claim 34 wherein the filter is a weighted synthesis filter.

36. The pitch predictor as claimed in Claim 34 wherein the predictor coefficients are selected as a function of the error signal.

37. The pitch predictor as claimed in Claim 34 wherein the adaptive codebook includes a lag factor.

38. The pitch predictor as claimed in Claim 37 wherein the lag factor is a function of the error signal.

39. The pitch predictor as claimed in Claim 33 wherein the vector quantization is conventional vector quantization.

40. The pitch predictor as claimed in Claim 33 wherein the vector quantization is product code vector quantization.

41. The pitch predictor as claimed in Claim 33 wherein the predictor coefficients are selected in a linear predictive analysis-by-synthesis manner.

42. A system for encoding speech signals, comprising:

an electronic device having a working memory and a digital processor; and

a pitch predictor executable in the working memory by the digital processor, the pitch predictor including:

at least one pitch predictor codebook having predictor coefficients; and

an adaptive codebook adjusted with a contribution from the adaptive codebook in combination with the predictor coefficients, the predictor coefficients being selected by searching the at least one pitch predictor codebook.

43. In a system having working memory and a digital processor, an apparatus for performing multi-tap pitch predictor vector quantization, the apparatus comprising:

at least one pitch predictor codebook having predictor coefficients; and

means for adjusting the adaptive codebook with a contribution from the adaptive codebook in combination with the predictor coefficients, the predictor coefficients being selected by searching the at least one pitch predictor codebook.

44. In a system having working memory and a digital processor, a method for producing a fixed codebook for a speech signal encoder, comprising:

filtering a target speech signal; and

forming entries in the fixed codebook of derived vector values indicating pulses in the filtered target speech signal.

45. The method as claimed in Claim 44 further including partitioning the filtered target speech signal into blocks.

46. The method as claimed in Claim 45 wherein the blocks are non-overlapping.

47. The method as claimed in Claim 45 wherein the blocks are overlapping.

48. The method as claimed in Claim 44 wherein the filtering is backward filtering.

49. The method as claimed in Claim 44 wherein the vector values are ternary vector values (1, 0, -1).

50. The method as claimed in Claim 44 wherein the vector values substantially indicate peak pulse positions in subvectors of the filtered target speech signal.

51. The method as claimed in Claim 44 further including considering non-contiguous positions in the filtered target speech signal to determine substantially peak pulse positions in the filtered target speech signal.

52. The method as claimed in Claim 44 wherein the derived vector values include sign and position information represented in bits.

53. The method as claimed in Claim 52 wherein the number of bits of the derived vector values includes a sign bit plus the number of bits representing, in binary, the number of locations within a subvector at which a peak pulse position is considered.

54. In a system having working memory and a digital processor, a fixed codebook for a speech signal encoder, comprising:

a filter filtering a target speech signal; and

entries in the fixed codebook of derived vector values indicating pulses in the filtered target speech signal.

55. The fixed codebook as claimed in Claim 54 wherein the filtered target speech signal is partitioned into blocks.

56. The fixed codebook as claimed in Claim 55 wherein the blocks are non-overlapping.

57. The fixed codebook as claimed in Claim 55 wherein the blocks are overlapping.

58. The fixed codebook as claimed in Claim 54 wherein the filter is a backward filter.

59. The fixed codebook as claimed in Claim 54 wherein the vector values are ternary vector values (1, 0, -1).

60. The fixed codebook as claimed in Claim 54 wherein the vector values substantially indicate peak pulse positions in subvectors of the filtered target speech signal.

61. The fixed codebook as claimed in Claim 54 in which non-contiguous positions in the filtered target speech signal are considered to determine substantially peak pulse positions in the filtered target speech signal.

62. The fixed codebook as claimed in Claim 54 wherein the derived vector values include sign and position information represented in bits.

63. The fixed codebook as claimed in Claim 62 wherein the number of bits of the derived vector values includes a sign bit plus the number bits representing, in binary, the number of locations within a subvector at which a peak pulse position is considered.

64. A system for encoding speech signals, comprising:

an electronic device having a working memory and a digital processor; and

a fixed codebook executable in the working memory by the digital processor, the fixed codebook including:

a filter filtering a target speech signal; and

entries in the fixed codebook of derived vector values indicating pulses in the filtered target speech signal.

65. In a system having working memory and a digital processor, an apparatus for producing a fixed codebook for a speech signal encoder, comprising:

means for filtering a target speech signal; and

means for forming entries in the fixed codebook of derived vector values indicating pulses in the filtered target speech signal.